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3.	Full name, address and postcode of the or of each applicant *(underline all surnames)*	PROFESSOR SELWYN EDGAR WRIGHT 22 Moor Lane Highburton HUDDERSFIELD HD8 0QS 7841497001
	Patents ADP number *(if you know it)*	
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4.	Tile of the invention	**"Active noise control system for reducting rapidly changing noise in unrestricted space"**
5.	Name of your agent *(if you have one)* "Address for service" in the United Kingdom to which all correspondence should be sent *(including the postcode)*	~~*Hulse & Co*~~ ~~*St James House, 8th Floor, Vicar Lane, Sheffield S1 2EX*~~ Page White & Farrer 54 Doughty Street London WC1N 2LS
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Drawing*(s)*	2 + 2 RW

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Any other documents *(please specify)*	

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Signature — Date

11 April 2002

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12. Name and daytime telephone number of person to contact in the United Kingdom — **Mr E A LONG** **(0114) 276 9381**

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ACTIVE NOISE CONTROL SYSTEM FOR REDUCTING RAPIDLY CHANGING NOISE IN UNRESTRICTED SPACE

BACKGROUND OF INVENTION

Conventional feedback adaptive cancellation systems using traditional transverse finite impulse response (FIR) filters, together with least mean square (LMS) adaptive algorithms, well known in the prior art, are slow to adapt to primary source changes. This makes them inappropriate for cancelling rapidly changing noise, including unpredictable noise such as speech and music. Secondly, the cancelling structures require considerable computational processing effort to adapt to primary source and plant changes, particularly for multi-channel systems.

In the case of cancelling single frequency sound, transverse FIR filters adapt with increasing speed (reduced time constant), as the number of taps (coefficients) is increased to an optimum value. Also for a single frequency or a group of frequencies the adaptive speed increases with the cancelling strength $\beta=\mu A^2$, where μ is the adaptive step size and A is the signal amplitude. Thus for a primary source with frequencies of various amplitudes, the lower amplitudes will adapt more slowly. If the signal is non-varying then the lower amplitude frequencies will adapt eventually. But for source frequencies varying in time the smaller amplitudes will never adapt completely, producing slow adaptation and signal distortion.

DISCLOSURE OF INVENTION

The invention is defined in claims1 and 2. Various preferred or optional features are defined in the appended sub-claims.

A method to increase the adaptive speed of transverse FIR filters to primary source changes is to divide the source spectrum into frequency passbands, where each passband has a separate FIR filter with its own μ made inversely proportional to A^2. Thus a faster and similar response, irrespective of signal amplitude, will be obtained in each passband.

This approach certainly increases the overall adaptive speed of conventional FIR filters, fairly evenly with spectrum amplitude, but unfortunately the adaptive speed is limited by a finite cancelling strength β. As β increases the stability bandwidth shrinks, its maximum value is given by the stability zero band width, as considered in Journal of Sound and Vibration (2001) 245(4).

The approach is therefore adequate for slowly changing primary sources such as unsteady periodic noise. Its disadvantage is intensive computation, requiring adaptive FIR filters and FIR passband filters for each band. Although the passband filters could be implemented into hardware reducing the computation burden.

Further disadvantages of the conventional transverse FIR adaptive systems are (i) basic instability, where the error sensor is permanently required and functioning to maintain stability (ii) deteriorated cancellation away from the error sensor and (iii) susceptiblility to environmental changes, through a large controlling propagation distance.

To implement a really fast response to source changes, including unpredictable noise, and avoiding the disadvantages listed above, the adaptive transverse FIR filter is removed all together and the primary source signal cancelled with a negative copy of itself, directly.

A time domain solution that gives virtually instantaneous response to primary source changes, and is computationally efficient, is to negate a copy of the primary source signal. Compensate for signal distortion caused through hardware implementation of the secondary cancelling system, align and match the resulting secondary wave with the primary wave.

This approach makes the cancellation process inherently stable, i.e. the error microphone is now needed only to set up the cancellation process. After the setting up, the cancellation is self-sustaining, without the use of the microphone, accept for all but severe environmental changes.

The resulting direct negative replica system, requiring no on-line adaptive FIR filters, has the following characteristics:

1) Secondary cancelling signal is 'copied' from the primary source using a primary sensing transducer (microphone or equivalent), suitably isolated from the secondary source (shielding and/or directional transducers) to prevent feedback between the two.

2) Secondary signal is negated in preparation for cancelling the primary signal.

3) The electromechanical plant (impulse) response I_{em}, which produces signal distortion, is neutralized/reduced by (i) physically altering the dynamic response of the dominant component - the sound transducer (loud speaker), (ii) mathematically modifying the net response of the system through adding the appropriate poles /zeros to the overall transfer function, (iii) measuring the impulse response of the system and inverting. The plant response includes essential components in the secondary sound cancelling path (computer A/D, D/A converters, aliasing/quatization filters, amplifiers and loudspeakers)

4) The physically modified plant response, and/or the neutralized plant response (inverted response I_{em}^{-1}convolved with the secondary plant signal), is used to drive the secondary loud speaker.

5) The resulting secondary acoustic wave is combined and aligned with the primary acoustic wave by appropriately positioning the secondary source downstream of the primary source in the direction of the wave propagation and error microphone. This facilitates a time advance relative to (along) the primary wave represented by the shift function $h(t+\tau_a)$.

6) Here the time advance $\tau_a = r_{ps}/c_o$, where τ_a is the wave propagation time between the primary and secondary sources, r_{ps} is the propagation distance between the sources and c_o is the propagation speed (speed of sound). The time advance is necessary to offset the delay represented through $h(t-\tau_r)$, where τ_r is the secondary path implementation/processing time retardation.

7) The cancellation is now dependent only on the difference distance between the primary and secondary sources, $r_{ps}= r_{pm}-r_{sm}$ (r_{pm} is the primary source microphone-error microphone distance), not on the secondary source-error microphone distance r_{sm}, as in the case of the conventional adaptive FIR process. As r_{ps} is considerable smaller than r_{sm}, the propagation space is much less vulnerable to environmental changes, such as fleeting reflections, than the conventional adaptive FIR method.

8) Acoustically the primary and secondary sources form a phase controlled dipole (PCD), as described in the Journal of Sound and Vibration (2001) **245** (4). Here the phase of the secondary source is adjusted to be out of phase with the primary sound field at the error microphone located downstream in successive alignment following the primary and secondary source. The resulting radiated acoustic field directivity (shadow shape) becomes progressively tripole (cardiod), dipole (figure of eight) and quadrupole (four leaf clover), as the difference between the primary and secondary source distance r_{ps} increases.

9) The PCD, in this direct negative replica case, uses the propagation distance r_{ps} for both the primary and secondary waves. This produces exact alignment between the waves, giving maximum shadow at all points along the wave from the primary source, in the direction of the error microphone. Whereas in the conventional adaptive FIR system, the propagation distance r_{pm} is used for the primary path and r_{sm} for the secondary path. This produces exact alignment only at the error microphone, giving a slight phase difference at all other points along the wave, progressively deteriorating the shadow with distance.

10) For digital systems, the control is implemented through samples generated by the sampling frequency f_n. The time advance τ_a is equivalent to a sample advance of:

$$n_a = \tau_a f_n = r_{ps} f_n / c_o \qquad (1)$$

11) The total sample delay (retardation) n_r is generated through (i) the inevitable secondary plant implementation time delay n_{imp}, including the inverse training delay n_{inv} and (ii) an adjustable sample delay n_b intentionally added, through a delay buffer, to fine tune the signal alignment, particularly through environmental changes. This gives a total retardation:

$$n_r = n_{imp} + n_b \qquad (2)$$

12) For a periodic wave, the secondary wave alignment with the primary wave at the secondary source (loud speaker) is given by:

$$N_p n_p - \Delta n = 0, \qquad \Delta n = n_r - n_a, \qquad n_p = T_p/T_n = f_n/f_p \qquad (3)$$

Where n_p is the number of samples in the period T_p of the primary wave of periodic frequency f_p and N_p is the period number that the primary wave is in advance of the secondary wave giving:

$$n_a = n_r - N_p n_p \qquad (4)$$

13) For a slowly changing periodic noise the system can be non-casual i.e. the delay τ_r can be longer than the advance τ_a, here only the periods need to be aligned i.e. N_p can be any integer. For unpredictable noise the signals must be casual and exactly aligned, the advance must balance the delay exactly, i.e. N_p=0, making

$$h(t+\tau_a)\, h(t-\tau_r) = h(t) \qquad (5)$$

14) The sample advance n_a is chosen through adjusting the distance between the primary and secondary source r_{ps}. The delay buffer n_b is then adjusted until $n_a = n_r$, giving a minimum error E at the error microphone.

15) The amplitude A of the secondary signal is adjusted to match that of the primary source signal giving a minimum error E at the error microphone.

16) The last two steps are successively repeated, manually or automatically, until the lowest minimum error E is achieved. This indicates that the secondary and primary signals are in alignment at the error microphone, and all points along the wave.

17) Shadows are formed at an angle α_B from the line joining the primary and secondary sources, from equation (1)

$$n_B = \Delta r_{ps}\, f_n/c_o, \qquad \Delta r_{ps} = r_{ps} - r_{ps}' = r_{ps}(1-\cos\alpha_B) \qquad (6)$$

where n_B is the buffer sample change, r_{ps}' is the propagation distance in the direction of the shadow minimum. Rearranging the above equation gives

$$\alpha_B = \cos^{-1}[(f_n r_{ps} - n_B c_o) / f_n r_{ps}] \qquad (7)$$

The shadow bending or rotation from the source axis, per n_B, therefore depends on the relative magnitude $f_n\, r_{ps}$ compared to c_o.

18) A single channel PCD cancelling system produces a narrow shadow. For practical systems requiring wide shadows, multi-channel (multi-speaker/multi error detector) systems are required. The primary source microphones, secondary cancelling sources and error microphones are successsively arranged in planes or arcs from the primary source and contained within shadow control angles forming well defined shadows.

19) Direct negative replica systems are fundamentally stable i.e. they do not require the error microphone to maintain cancelling stability. For a non-changing plant the error microphone can be dispensed with after the initial setting up to produce minimum error (sound).

20) Thus each channel in these multi-channel systems can be set up independently, requiring no inter-channel coordination. Of course a multi-channel computer coordinated system should always out-perform a set of independent systems, particularly for operation in significantly changing environments.

To obtain minimum distortion of the cancelling process, resulting in maximum cancellation, it is important to obtain neutralization of the secondary plant response. This can be obtained through an accurate measurement of the inverse of the electromechanical plant impulse response $(I_{em}^{*})^{-1}$ as follows.

Measuring an estimate of $(I_{em}^{*})^{-1}$, in the time domain, directly in series with the actual I_{em}, on or off line, using a white noise training signal. Care is needed in performing direct inverse estimates, as inverted functions are potentially unstable. For example proper functions (functions with more poles than zeros) become improper functions when inverted.

More seriously, 'unstable' zeros lying outside the unit circle in the Z domain (non-minimum phase functions) become unstable poles, when inverted. For non minimum phase functions a large delay z^{-n} is required for the adaptive process to converge effectively.

An alternative method, that avoids the instability problem and requires a small delay, is to obtain the inverse directly from the impulse response. An estimate I_{em}^{*} is measured in parallel with the actual I_{em}, on or off line, using a white noise training signal. The spectrum amplitude B and phase θ are then obtained through performing the Discrete Fourier Transform or swept spectrum or equivalent on I_{em}* thus:

$$\mathrm{DFT}(I_{em}^{*}) = \Sigma\,(Be^{j\omega\theta}) \qquad (8)$$

The inverse is then obtained by simply inverting B and negating θ, and then reassembling back into the time domain, thus:

$$(\mathrm{DFT})^{-1} = \Sigma\,(B^{-1}e^{-j\omega\theta}) = (I_{em}^{*})^{-1} \qquad (9)$$

The negative replica cancelling system is basically instantaneous to the response of primary source changes, as a copy of the primary source signal is passed directly through the secondary source system to the cancelling loud speaker. Apart from the convolution, there are no computational demanding processes either. A simple phase and amplitude error adjustment is affected using a simple delay buffer and amplitude regulator

DESCRIPTION OF DRAWINGS

FIG 1 is a block diagram of the multi-bandpass, variable μ, transverse FIR adaptive filter
FIG 2 is a block diagram of the direct negative replica , non- FIR adapting filter
FIG 3 is a block diagram of three possible multi-channel configurations using the direct negative replica technique.

DESCRIPTION OF INVENTION

Referring to FIG 1, the primary noise to be cancelled propagates from the primary source **1** along the propagation path **2** to the error transducer (microphone) **3**. The primary noise signal is measured, in close proximity to the primary source **1,** using a primary transducer (microphone) **4**. The primary signal is fed through pass-band 1 filter **5a,** and then through a conventional finite impulse response FIR 1 filter **6a,** to the secondary transducer (loud speaker) 7. The loud speaker generates the secondary noise that propagates through the secondary propagation space **8** to the detection microphone **3**.

The output from pass-band 1 filter **5a,** is passed also through a conventional plant estimate **9a** into a conventional least mean squared LMS 1 algorithm **10a**. Also fed into the LMS 1 algorithm **10a** is the error signal E from the error microphone **3** and the adaptive step size **11a,** automatically calculated from the band-pass 1 output level. The output from the LMS algorithm **10a** then controls the FIR 1 filter **6a** adaptive process to drive that part of the error signal E in its pass-band 1 to a minimum.

Similarly, the output from the primary microphone **4** is passed into pass-band 2 filter **5b,** through the FIR 2 filter **6b** into the secondary loud speaker **7.** The loud speaker generates the secondary noise that propagates through the secondary propagation space **8** to the detection microphone **3**. The output from pass-band 2 filter **5b** is passed through the same plant estimate **9b,** then into the LMS 2 algorithm **10b,** together with the error signal E from the error microphone **3** and the output from the automatic adaptive step size **11b,** whose size is determined by the output from pass-band 2 filter **5b**. The output from the LMS algorithm **10b** then controls the FIR 2 filter **6b** adaptive process to drive the error signal in its pass-band to a minimum. To extend the total frequency range or reduce the spectrum density per passband, additional 'n' pass-band adaptive systems, each minimizing the spectrum amplitude in each of its pass-bands, can be added.

As the adaptive speed is proportional to the signal amplitude 'A' squared times the adaptive step size μ in each pass-band, then if the step size is reduced proportional to the signal amplitude squared, automatically, then equally high adaptive speeds will be obtain for all frequencies and amplitudes within each pass-band. This increases the overall

adaptive speed and reduces the spectrum distortion compared with a conventional transverse FIR filter. However, there is a limit to the product size μA^2 limiting the adaptive process to slowly changing periodic noise.

Referring to FIG 2, to increase the response to rapidly changing primary sources, to avoid the disadvantages of conventional adaptive FIR filters, listed and discussed earlier, and to reduce the computation effort, box **12** in FIG1 is replaced with box **17** in FIG 2.

To deal with the cancellation of arbitrary noise (non-periodic unpredictable noise) the process is described generally in the time domain. Again to generate the secondary cancelling signal, a copy of primary source signal is measured using the primary microphone **4**.

The output from the primary microphone **4** is negated **13**, and then convolved with the plant neutralization inverse **14**, which removes the signal distortion produced by the cancelling system hardware. The plant inverse **14** can be measured directly in series with the plant or determined from its impulse response measured in parallel with the plant. The inverse can be obtained also through the frequency domain, as described in detail previously. The secondary signal is then passed through an adjustable delay buffer **15**, amplitude control **16**, and to the secondary loud speaker **7**, where the resulting sound propagates through the secondary propagation space **8**, to the error microphone **3**.

The secondary signal is aligned with the primary signal using the sample delay buffer n_b **15**, which is adjusted to give minimum error E at the error microphone **3**. The amplitude of the secondary signal is matched to that of the primary signal by adjusting the amplitude **16**, to give a minimum error at the error microphone **3**. The amplitude A and the delay n_b are then successively adjusted until a steady minimum error is achieved at the error microphone **3**, manually or automatically.

For cancelling steady periodic noise the periods need only to be aligned (N_p integer in equation 4). For unpredictable noise the secondary signal needs to be aligned exactly with the primary signal (N_p=0). This is accomplished by adjusting the propagation distance sample advance between the secondary loudspeaker and the primary source n_a to that of the computation delay n_r, in equation 4.

As a single (PCD) channel generates only a narrow cancellation region (shadow), multi channel systems (multi secondary sources and error detectors) are required to generate a practical shadow over a wide well defined angle. FIG 3 shows three possible configurations.

Figure 3(a) shows the configuration for a small or large in-phase primary source **1** generating a shadow over a wide shadow angle. Here a single primary microphone **4** is sufficient to drive all the secondary sources **7**. A single error microphone **3** is sufficient to adjust each channel, one at a time, at each of the angle positions, as indicated with the dotted outline. With in the adjustable control boxes **17** are the control elements shown in the chain dotted box **17** in Figure (2). The secondary sources **7** and error detectors **3** are

arranged in successive planes or arcs from the primary source and contained within control angles **18** forming the shadow angle.

Figure 3(b) is a configuration for an out of phase primary source **1** (for example containing modal distributions). Here separate primary microphones **4** are used to measure the local sound across the primary source and drive each channel separately, making them self-contained units. Each unit consists of a primary microphone **4**, control system **17**, and loud speaker **7**. Again only a single error microphone is used in turn, at each angular position, to minimize the error signal for each channel, one at a time.

Finally, Figure 3(c) shows an array of self-contained units **4, 17** and **7** and an array of permanent error microphones **3** shown in full line. Each of the control systems **17**, and each of the error microphones **3** are linked to a computer **19**. The control systems **17** are adjusted automatically through the computer **19** to produce a minimum collective error at the error microphones **3**. All three configurations are capable of shadow angle rotation through appropriate adjustment of n_B, r_{ps} or f_n.

CLAIMS

What is claimed is:

1. An active noise control system comprising:
primary sensor means for detecting the primary acoustic wave from a primary source and providing an output signal indicative of the primary acoustic wave;
sound producing means that receives a processed output signal from the primary sensor means and produces a secondary acoustic wave used to cancel the unwanted primary acoustic wave;
error sensor means that senses the difference (error) between the primary and secondary acoustic waves and provides an output signal;
signal processing means that processes the output from the primary sensor means and outputs to the sound producing means according to the output from error sensor means, minimizing the said error producing a region of reduced sound (acoustic shadow);
characterized in that the response is increased to cancel unsteady periodic noise where the output from said primary sensor means is filtered through a plurality of pass-bands, each pass-band feeds a conventional adaptive finite impulse response (FIR) filter with a conventional least mean squared (LMS) adaptive algorithm, wherein its adaptive step size is made proportional to the amplitude squared within each pass-band, the adaptive process for each pass-band proceeds until the error is a minimum at the said error sensor means.

2. An active noise control system comprising:
primary sensor means for detecting the primary acoustic wave from a primary source and providing an output signal indicative of the primary acoustic wave;
sound producing means that receives a processed output signal from the primary sensor means and produces a secondary acoustic wave used to cancel the unwanted primary acoustic wave;
error sensor means that senses the difference (error) between the primary and secondary acoustic waves and provides an output signal;
signal processing means that processes the output from the primary sensor means and outputs to the sound producing means according to the output from error sensor means, minimizing the said error producing a region of reduced sound (acoustic shadow);
characterized in that the response is made instantaneous to cancel unpredictable noise through avoiding source adaptive processes, providing a time advance facility and compensating for plant distortions, where the signals from said primary sensor means are negated, convolved with the plant response inverse, aligned and matched in amplitude with the primary wave until the error is a minimum at the said error sensor means.

3. The control system of claim 2 wherein the inevitable secondary signal processing sample delay n_r is offset by a signal advance n_a, produced by positioning the sound producing means downstream of the primary source wave until the secondary cancelling wave is in advance of the primary wave i.e. $n_a > n_r$

4. The control system of claim 2 wherein the secondary cancelling wave is exactly aligned, $n_a = n_r$, and matched in amplitude with the primary wave to be cancelled through an adjustable circular buffer sample delay number n_b and an amplitude adjuster A, respectively, implemented in computer code.

5. The control system of claim 2 wherein the distance between said primary sensor means and said sound producing means r_{ps}, which controls the cancellation process, is made much smaller than the distance between said sound producing means and error detecting means r_{sm}, making the cancellation process less vulnerable to environmental changes (than in claim 1 and prior art, where the larger distance between said sound producing means and error detecting means r_{sm} controls the cancellation process).

6. The control system of claim 2 wherein the secondary control path r_{ps} is made the same as the primary path, enabling the secondary cancelling signal to be completely aligned at all points along the primary signal, producing a uniform shadow along the wave, (whereas the control system in claim 1 and prior art, the secondary control path r_{sm} is not identical to the primary path r_{pm} allowing exact phase with the primary wave only at the error detecting means, thus producing a deteriorating shadow with distance along the wave).

7. The control system of claim 2, wherein the instability of inverting minimum phase functions, is avoided through obtaining the spectrum amplitude and phase of the plant impulse response, inverting the amplitude and negating the phase and then reassembling the time domain inverse function.

8. The control system of claim 2, wherein the acoustic shadows can be rotated from the line joining the primary and secondary sources, the rotation angle depending on the buffer delay change, sampling frequency, primary-secondary source distance and speed of sound.

9. The control system of claim 2 wherein a plurality of said primary sensor means, sound producing means and error sensor means are in successive alignment planes or arcs from the primary source, and contained with in shadow control angles.

10. The control system of claim 9 wherein each individual primary sensor means, sound producing means and errors sensing means are considered as a linear unit, where each sound producing means is individually adjusted to produce a minimum error at each individual error sensing means.

11. The control system of claim 9 wherein primary sensor means, sound producing means and errors sensing means are considered as a whole, where the sound producing means are collectively adjusted, through computer co-ordination, to produce a total collective minimum error at the error sensing means.

ABSTRACT

ACTIVE NOISE CONTROL SYSTEM FOR REDUCTING RAPIDLY CHANGING NOISE IN UNRESTRICTED SPACE

The adaptive speed to changes in a primary source noise **1** is increased through a noise detecting means **4** feeding parallel multi-passband means **5** and multi-transverse adaptive filter means **6**, where each adaptive filter has its own individual adaptive step size means **11** adjusted automatically according to the signal strength at each passband output. The output from each of the multi-adaptive filter means **6** drives a secondary cancelling source generating means **7** where each multi-adaptive filter means **5** is automatically adjusted to produce minimum sound in its passband at an error detecting means **3**. Alternatively, the output from the noise detecting means **4** is negated through a negation means **13**, passed through a plant neutralization inverse means **14** before driving the secondary source generating means **7**. The secondary source output is aligned and match in amplitude to that of the primary source **1**, through a delay buffer means **15** and an amplitude regulator means **16**, which are adjusted successively until the output at the error detector means **3** is a minimum.

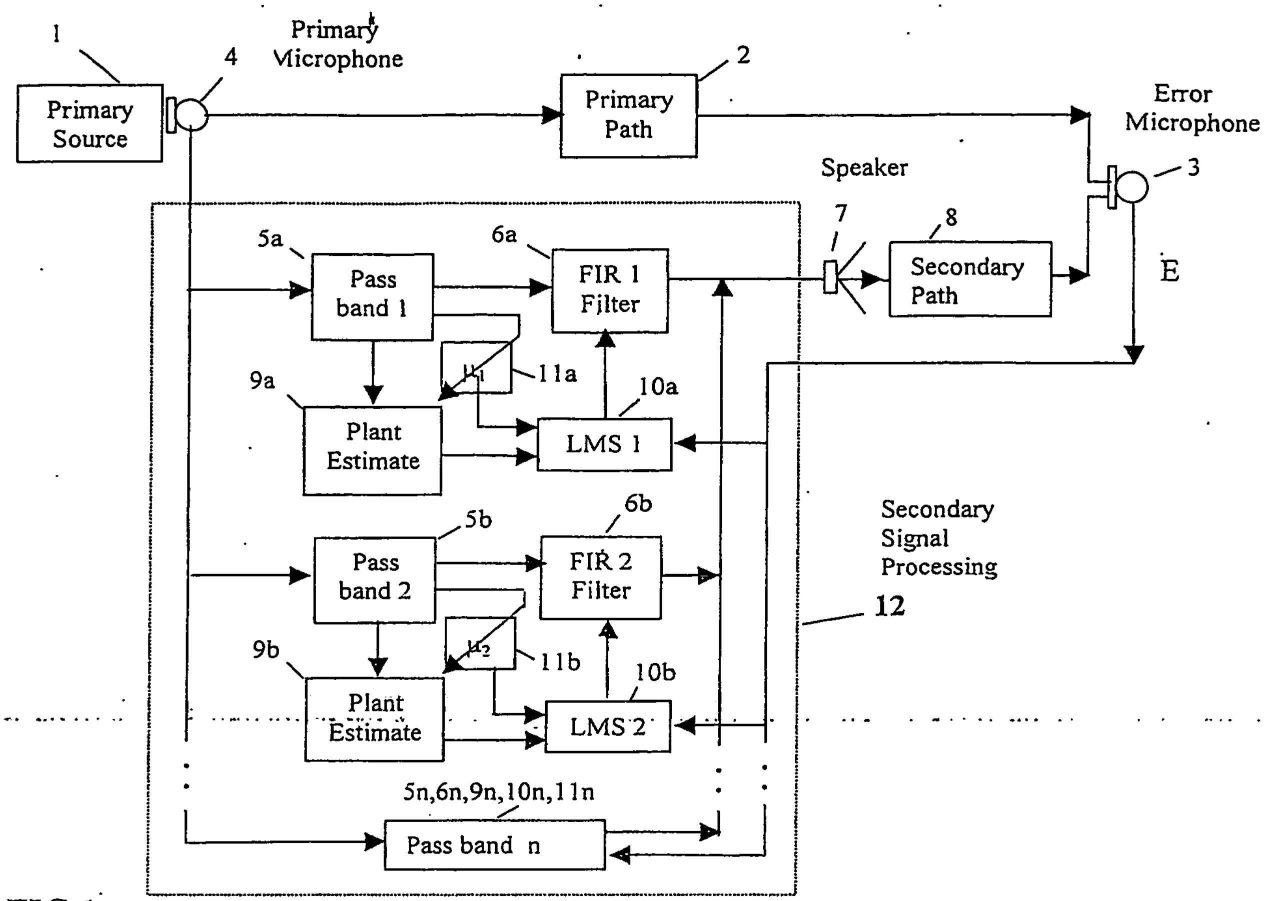
1
Primary Source
4
Primary Microphone
Primary Path
2
Error Microphone
3
Speaker
7
8
Secondary Path
E
5a
6a
Pass band 1
FIR 1 Filter
μ1
11a
9a
10a
Plant Estimate
LMS 1
6b
5b
Pass band 2
FIR 2 Filter
Secondary Signal Processing
12
9b
μ2
11b
10b
Plant Estimate
LMS 2
5n,6n,9n,10n,11n
Pass band n

FIG 1

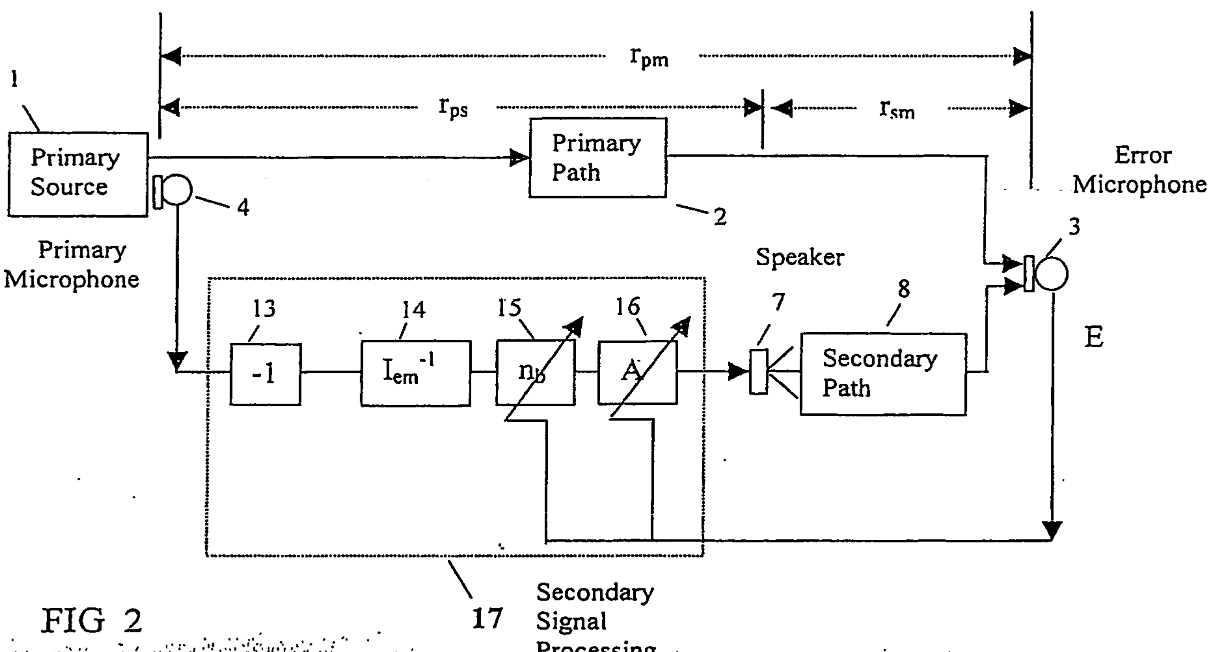
1
rpm
rps
rsm
Primary Source
Primary Path
4
2
Error Microphone
3
Primary Microphone
Speaker
13
14
15
16
7
8
-1
Iem-1
nb
A
Secondary Path
E
17
Secondary Signal Processing

FIG 2

03|01565
PJT